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Murali

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(54) **DE-EMPHASIS FILTERING AUDIO SIGNALS IN RESPONSE TO COMPOSITE CONTROL SIGNAL**

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H04B 1/04 (2006.01)
H04L 27/12 (2006.01)

(52) **U.S. Cl.**
CPC ***H04B 1/034*** (2013.01); ***H04B 1/0475*** (2013.01); ***H04L 27/12*** (2013.01)

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CPC H04B 1/04; H04B 1/0475; H04B 1/02; H04B 1/00; H04B 1/034; H04B 1/1027; H04B 14/006; H04B 14/005; H03F 1/3247; H03F 1/0277; H03F 1/0205; H03F 3/211; H03F 3/24; H03F 2201/3233; H04L 27/368; H04L 25/03343; H04L 27/20; H04L 27/12; H03C 5/00; H03C 3/02; H03C 3/18
USPC 381/16, 2, 13, 10, 3; 700/94; 455/114.2, 455/112
See application file for complete search history.

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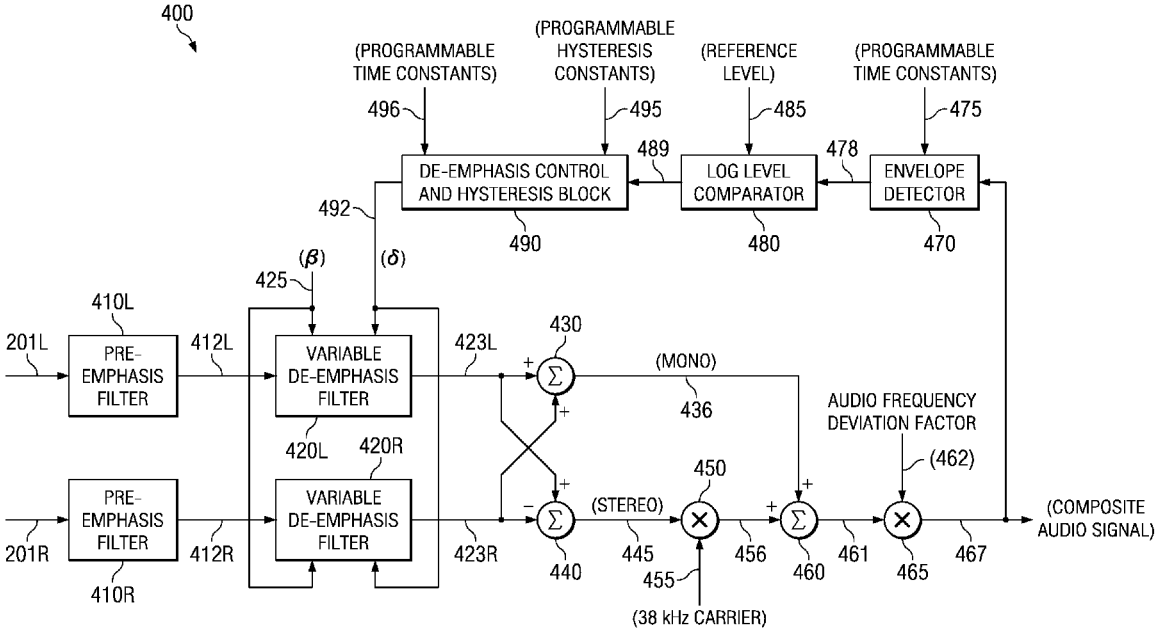
(57) **ABSTRACT**

Digital values representing an audio signal are formed in a FM transmitter. The deviation of the amplitude of the audio signal as represented by the digital values, from a reference amplitude level, is measured. The audio signal is filtered based on the deviation measured by processing the digital values, with the filtering designed to filter different frequency components of the audio signal with different respective magnitudes. Another set of digital values representing the filtered audio signal is generated, and used to frequency modulate a carrier signal to generate a FM signal. Over-modulation of the FM signal is thereby controlled.

12 Claims, 5 Drawing Sheets



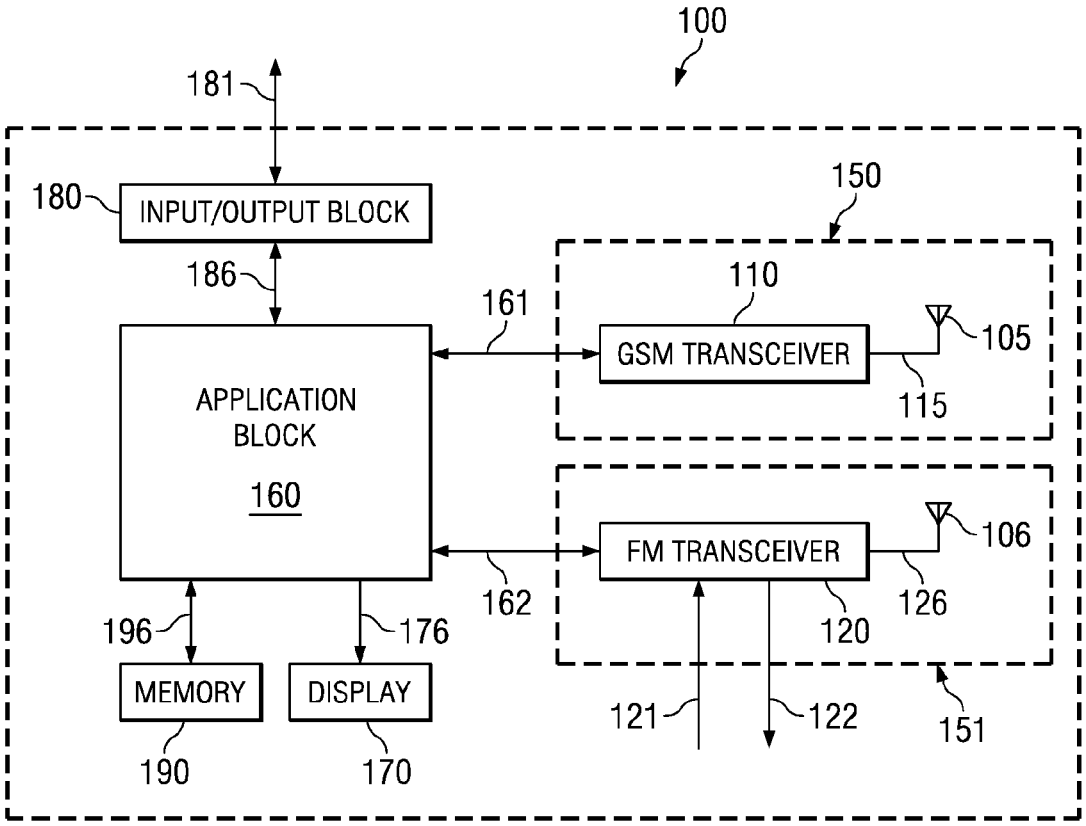

100
181
180
INPUT/OUTPUT BLOCK
186
150
110
161
GSM TRANSCEIVER
105
115
APPLICATION BLOCK
160
FM TRANSCEIVER
106
126
162
120
196
176
MEMORY
DISPLAY
121
122
151
190
170

FIG. 1

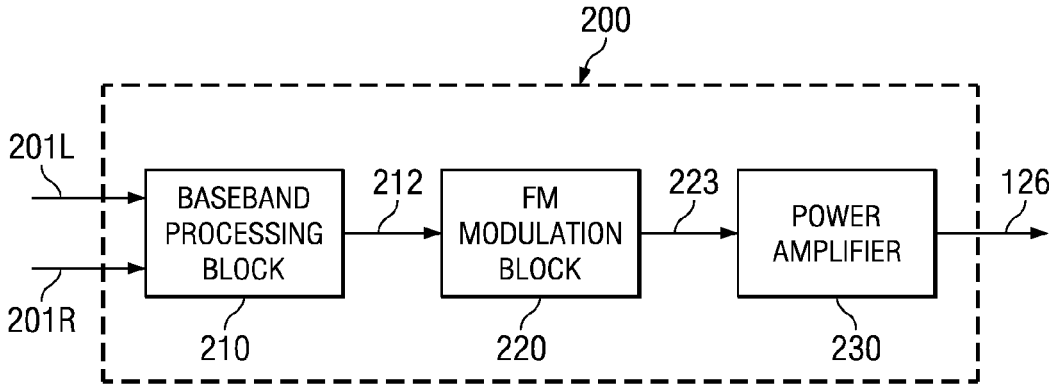

200
201L
201R
BASEBAND PROCESSING BLOCK
212
FM MODULATION BLOCK
223
POWER AMPLIFIER
126
210
220
230

FIG. 2A

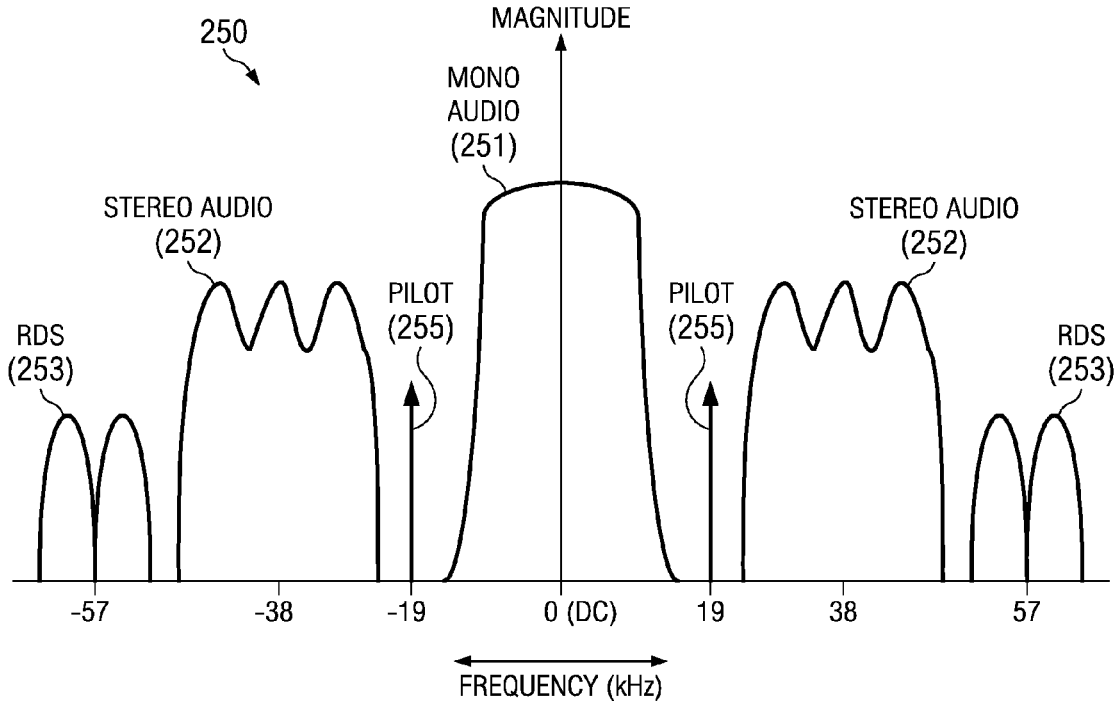

250
MAGNITUDE
MONO AUDIO (251)
STEREO AUDIO (252)
STEREO AUDIO (252)
PILOT (255)
PILOT (255)
RDS (253)
RDS (253)
-57
-38
-19
0 (DC)
19
38
57
FREQUENCY (kHz)

FIG. 2B

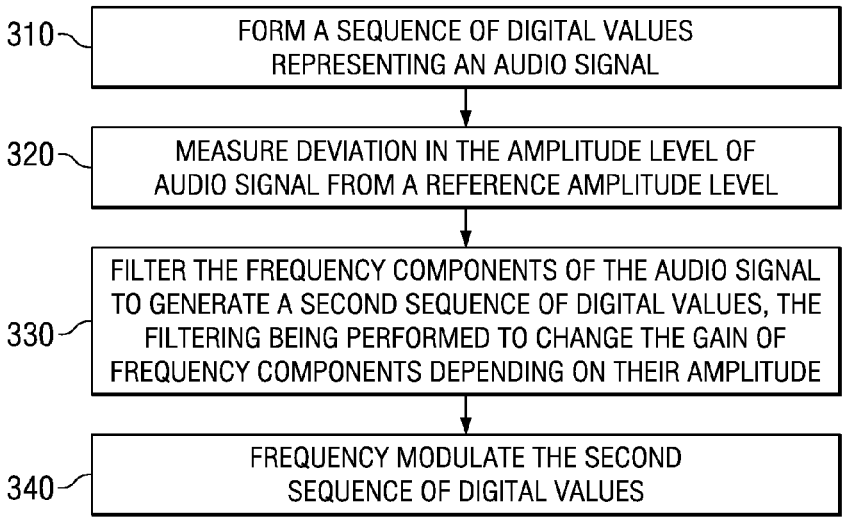

310
FORM A SEQUENCE OF DIGITAL VALUES REPRESENTING AN AUDIO SIGNAL
320
MEASURE DEVIATION IN THE AMPLITUDE LEVEL OF AUDIO SIGNAL FROM A REFERENCE AMPLITUDE LEVEL
330
FILTER THE FREQUENCY COMPONENTS OF THE AUDIO SIGNAL TO GENERATE A SECOND SEQUENCE OF DIGITAL VALUES, THE FILTERING BEING PERFORMED TO CHANGE THE GAIN OF FREQUENCY COMPONENTS DEPENDING ON THEIR AMPLITUDE
340
FREQUENCY MODULATE THE SECOND SEQUENCE OF DIGITAL VALUES

FIG. 3

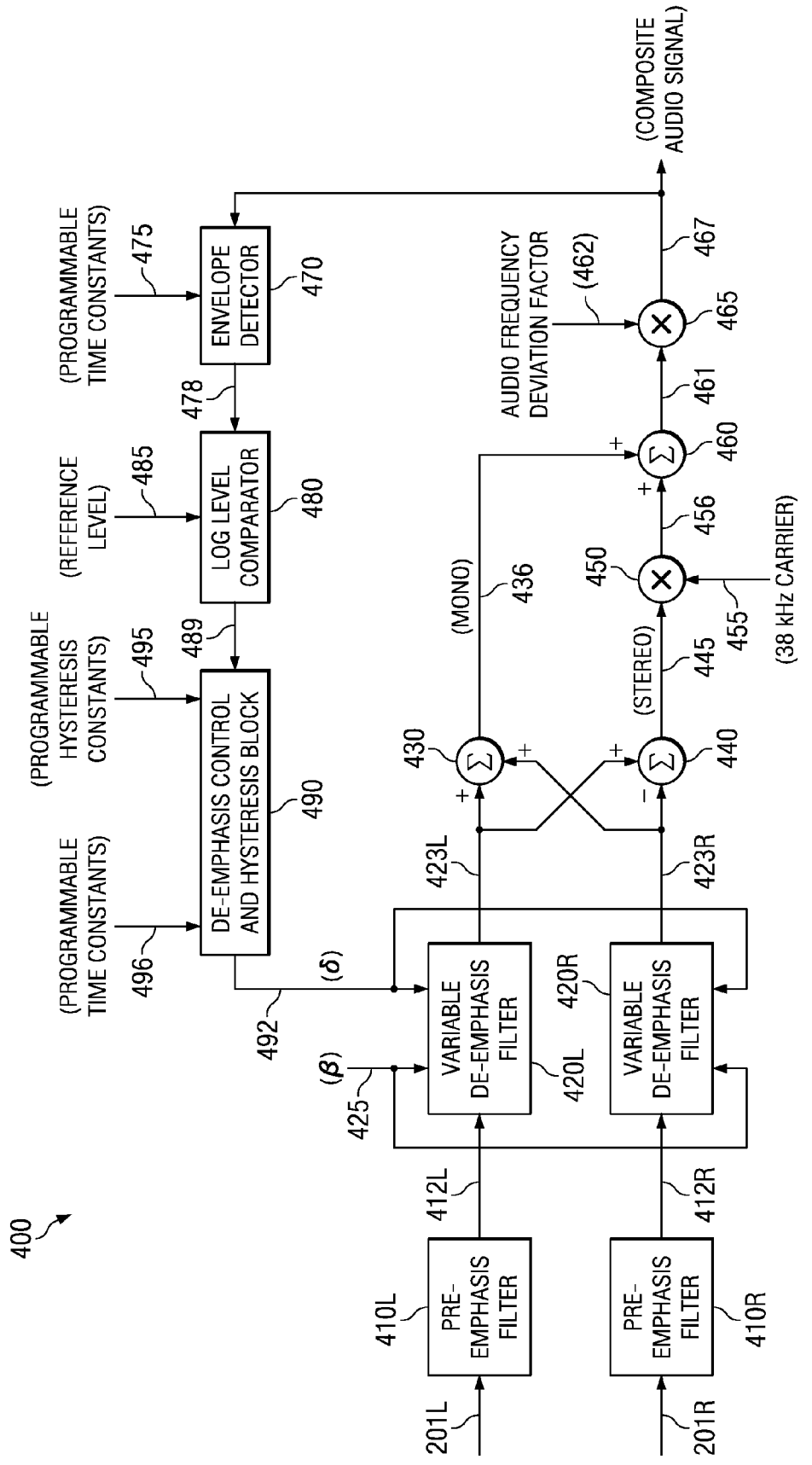

400
201L
201R
PRE-EMPHASIS FILTER
410L
410R
412L
412R
VARIABLE DE-EMPHASIS FILTER
420L
420R
425
(β)
492
(δ)
DE-EMPHASIS CONTROL AND HYSTERESIS BLOCK
490
(PROGRAMMABLE TIME CONSTANTS)
496
(PROGRAMMABLE HYSTERESIS CONSTANTS)
495
LOG LEVEL COMPARATOR
480
(REFERENCE LEVEL)
485
489
ENVELOPE DETECTOR
470
(PROGRAMMABLE TIME CONSTANTS)
475
478
423L
423R
430
440
(MONO)
436
(STEREO)
445
450
455
(38 kHz CARRIER)
456
460
461
AUDIO FREQUENCY DEVIATION FACTOR
(462)
465
467
(COMPOSITE AUDIO SIGNAL)

FIG. 4

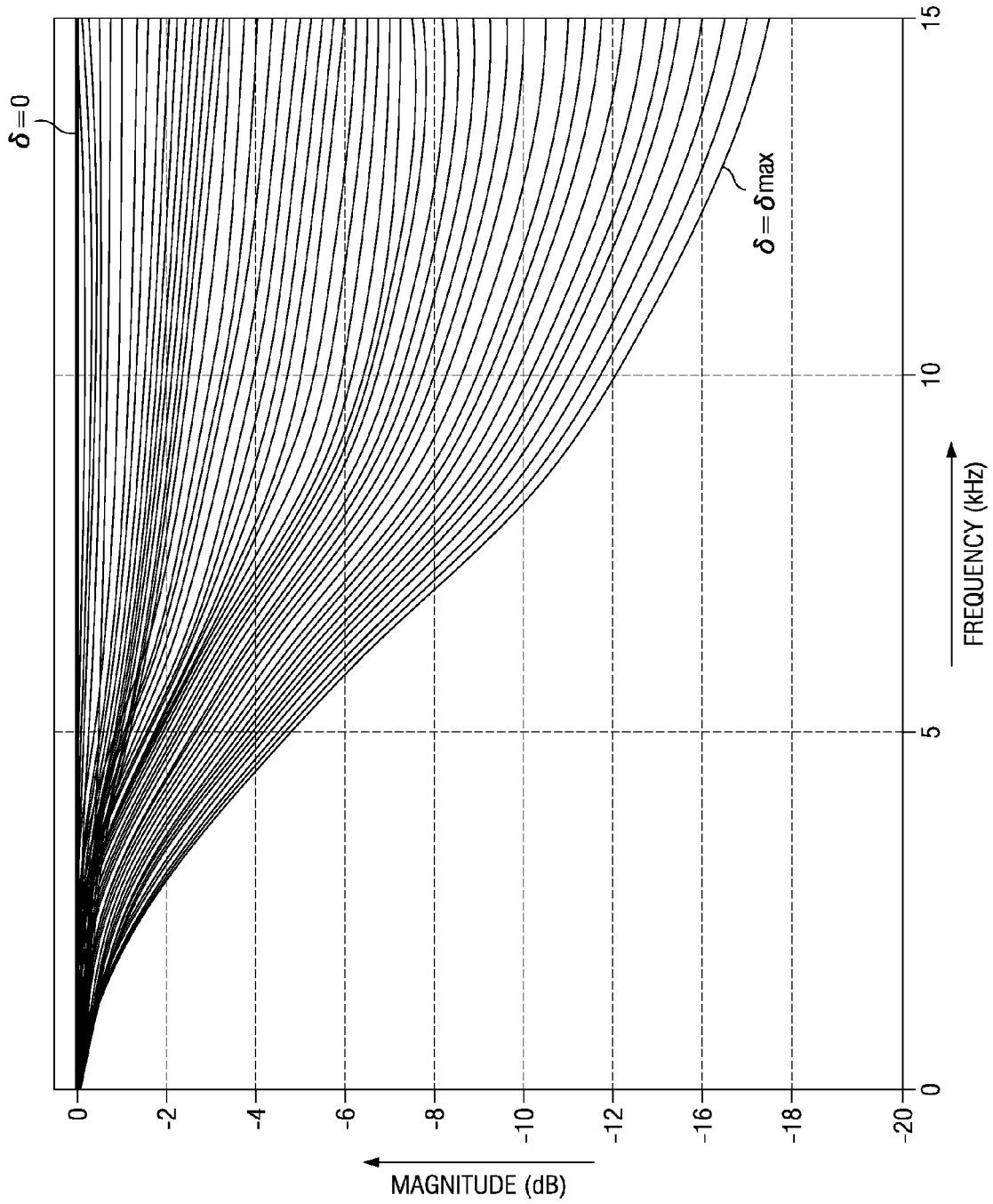

δ=0
δ=δmax
0
-2
-4
-6
-8
-10
-12
-16
-18
-20
MAGNITUDE (dB)
0
5
10
15
FREQUENCY (kHz)

FIG. 5

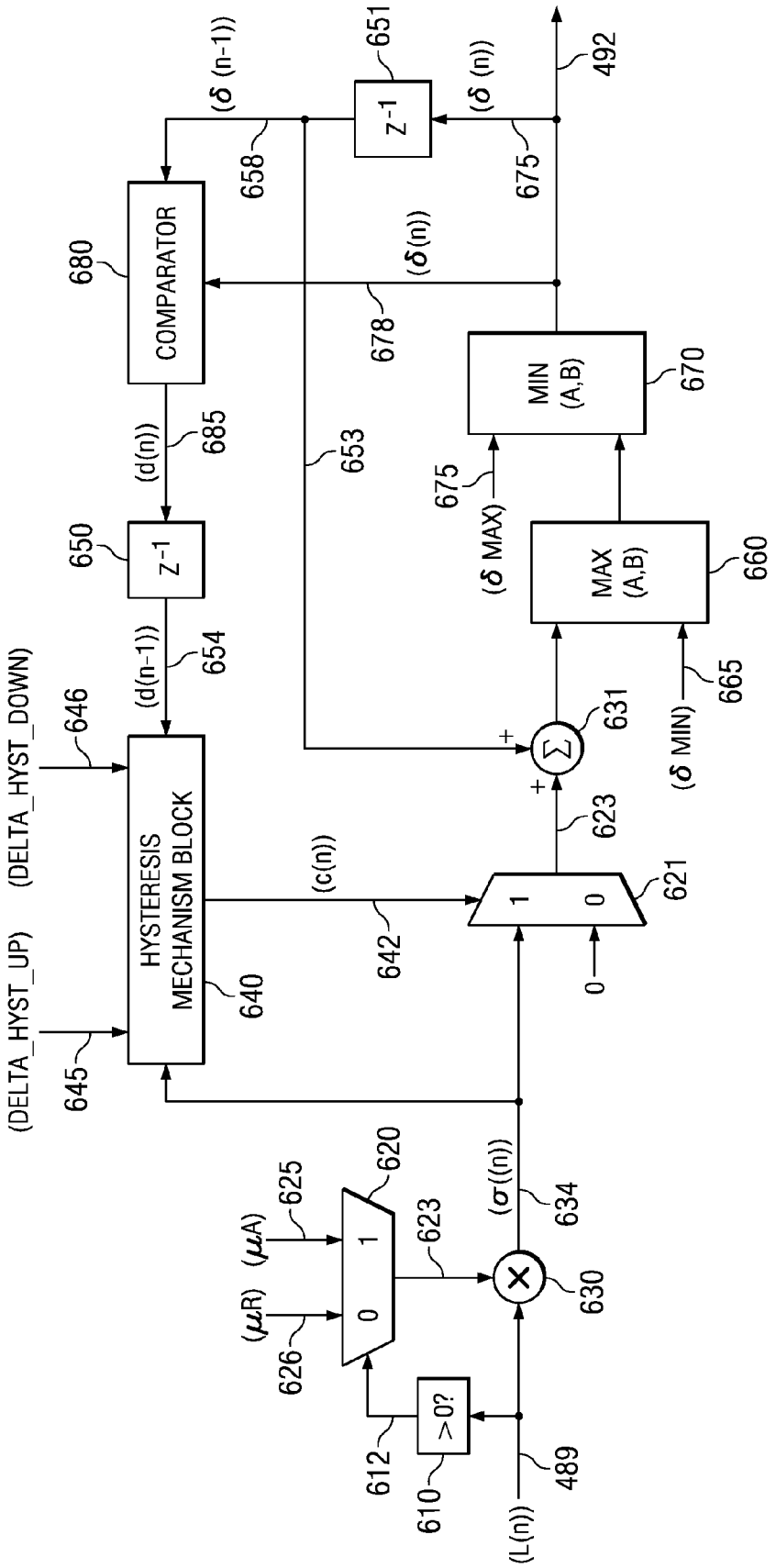

(δ (n-1))
651
Z-1
(δ (n))
492
658
675
680
COMPARATOR
(δ(n))
678
MIN (A,B)
670
653
(d(n))
685
675
(δ MAX)
650
Z-1
MAX (A,B)
660
(d(n-1))
654
(DELTA_HYST_DOWN)
646
631
665
(δ MIN)
623
(c(n))
621
HYSTERESIS MECHANISM BLOCK
642
(DELTA_HYST_UP)
645
640
0
(σ((n))
634
620
625
(μA)
623
630
(μR)
626
612
>0?
610
489
(L(n))

FIG. 6

DE-EMPHASIS FILTERING AUDIO SIGNALS IN RESPONSE TO COMPOSITE CONTROL SIGNAL

RELATED APPLICATION(S)

The present application claims the benefit of co-pending India provisional application serial number: 583/CHE/2009, entitled: "Method and Apparatus for Adaptive Audio De-emphasis Based Over-modulation Protection in FM Transmitters", filed on 16 Mar. 2009, naming Texas Instruments Inc. (the intended assignee) as the Applicant, and naming the same inventor as in the present application as inventor, and is incorporated in its entirety herewith.

BACKGROUND

1. Technical Field

Embodiments of the present disclosure relate generally to FM transmitters, and more specifically to controlling over-modulation in FM transmitters.

2. Related Art

Frequency Modulation (FM) refers to modulation of the frequency of a carrier signal in proportion to the instantaneous amplitude of a message signal. The bandwidth of a FM signal increases with increase in message signal amplitude.

FM transmission/broadcast standards specify the allowable center frequencies for the carrier signals and a maximum allowable bandwidth that can be occupied by the frequency modulated signal. Over modulation refers to a situation in which the bandwidth occupied by a FM signal exceeds the maximum allowable bandwidth, and accordingly the FM signal is said to be over-modulated.

Over-modulation may have undesirable effects such as interference with FM signals on adjacent FM bands. It is, therefore, desirable that over-modulation be controlled in FM transmitters.

SUMMARY

This Summary is provided to comply with 37 C.F.R. §1.73, requiring a summary of the invention briefly indicating the nature and substance of the invention. It is submitted with the understanding that it will not be used to interpret or limit the scope or meaning of the claims.

Digital values representing an audio signal are formed in a FM transmitter. The deviation of the amplitude of the audio signal as represented by the digital values, from a reference amplitude level, is measured. The audio signal is filtered based on the deviation measured by processing the digital values, with the filtering designed to filter different frequency components of the audio signal with different respective magnitudes. Another set of digital values representing the filtered audio signal is generated, and used to frequency modulate a carrier signal to generate a FM signal.

Several aspects are described below with reference to examples for illustration. It should be understood that numerous specific details, relationships, and methods are set forth to provide a full understanding of the invention. One skilled in the relevant art, however, will readily recognize that the invention can be practiced without one or more of the specific details, or with other methods, etc. In other instances, well-known structures or operations are not shown in detail to avoid obscuring the features of the invention.

BRIEF DESCRIPTION OF THE VIEWS OF DRAWINGS

FIG. **1** is a block diagram of an example device in which several embodiments can be implemented.

FIG. **2**A is a block diagram illustrating the details of portion of a FM transmitter in an embodiment.

FIG. **2**B is a diagram illustrating the frequency spectrum of a composite message signal according to FM standards.

FIG. **3** is a flow diagram illustrating the manner in which over-modulation in FM transmitters is controlled in an embodiment.

FIG. **4** is a block diagram illustrating partial details of a baseband processing block used in a FM transmitter, in an embodiment.

FIG. **5** is a graph depicting frequency response curves of a variable de-emphasis filter used in a FM transmitter, in an embodiment.

FIG. **6** is a block diagram illustrating the implementation blocks of a de-emphasis control and hysteresis block used in a FM transmitter, in an embodiment.

DETAILED DESCRIPTION

Various embodiments are described below with several examples for illustration.

1. Example Device

FIG. **1** is a block diagram of an example device in which several aspects of the present disclosure can be implemented. The block diagram shows mobile phone **100**, which is in turn shown containing GSM (Global System for Mobile Communication) block **150**, FM block **151**, application block **160**, display **170**, input/output (I/O) block **180**, and memory **190**. The components/blocks of mobile phone **100** in FIG. **1** are shown merely by way of illustration. However, mobile phone **100** may contain more or fewer components/blocks. Further, while in the examples below, over-modulation control techniques are described with respect to FM transceivers in mobile phones, the techniques can be applied in the context of FM transceivers in other environments as well.

GSM block **150** is shown containing GSM transceiver **110** and transmit antenna **105**. FM block **151** is shown containing FM transceiver **120** and transmit antenna **106**. GSM block **150**, and FM block **151** may contain respective receive antennas and filters as well, but are not shown in FIG. **1**.

Blocks **110** and **120** may be implemented, for example, as separate integrated circuits (IC), or implemented within a same IC. Typically, antennas **105** and **106** (as well as one or more of components such as filters assumed to be contained within blocks **110** and **120**) of FIG. **1** are mounted on a printed circuit board (PCB), with corresponding PCB trace providing the electrical connectivity represented by paths **115** and **126**.

GSM block **150** operates to provide wireless telephone operations, with GSM transceiver **110** containing receiver and transmitter sections to perform the corresponding receive and transmit functions. FM block **151** operates to transmit and receive FM audio signals according to FM broadcast/receive standards.

Application block **160** may contain corresponding hardware circuitry (e.g., processors), and operates to provide various user applications provided by mobile phone **100**. The user applications may include voice call operations, data transfers, providing positioning information, etc. Application block **160** may operate in conjunction with blocks **150** and **151** to provide such features, and communicates with the respective blocks **150** and **151** via paths **161** and **162** respectively.

Display **170** displays image frames in response to the corresponding display signals received from application block **160** on path **176**. The images may be generated by a camera provided in mobile phone **100**, but not shown in FIG. **1**. Display **170** may contain memory (frame buffer) internally for temporary storage of pixel values for image refresh pur-

poses, and may be implemented, for example, as a liquid crystal display screen with associated control circuits. Input/Output (I/O) block **180** provides a user with the facility to provide inputs via path **181**, for example, to dial numbers. In addition, I/O block **180** may provide on path **181** outputs that may be received via application block **160** via path **186**. Such outputs may include data, voice, images etc. I/O block **180** communicates with application block **160** via path **186**.

Memory **190** stores program (instructions) and/or data (provided via path **196**) used by applications block **160**, and may be implemented as RAM, ROM, flash, etc, and thus contains volatile as well as non-volatile storage elements, and represents a computer (or a machine) readable medium.

FM transceiver **120** may include FM transmitter as well as FM receiver sections, and receives audio signals (for example from microphones, not shown) on path **121**. Path **121** is deemed to contain at least two separate paths, one each for left and right audio channels of a stereo audio signal received on path **121**. FM transceiver **120** provides audio signals as output on path **122**, which may, for example, be connected to a speaker (not shown). Alternatively, FM transceiver **120** may receive stereo audio signals in digital form via block I/O **180** and application block **160** on path **162**, and send received (and demodulated) audio signals on path **162** to I/O block **180** via application block **160** and path **162**.

The implementation of a transmitter section contained in FM transceiver **120** is described next with respect to FIG. **2**. FIG. **2**A is a block diagram illustrating the details of FM transmitter **200** in an embodiment. FM transmitter **200** is shown containing baseband processing block **210**, FM modulation block **220** and power amplifier **230**.

Baseband processing block **210** is shown connected to input paths **201**L and **201**R on which respective channels of a stereo audio signal (referred to for convenience as audio signal **201**) may be received. Baseband processing block **210** may combine signals **201**L and **201**R (left and right channels representing components of stereo audio signal **201**) into a composite message signal, sent on path **212**. The composite message signal may be formed according to FM broadcast standard specifications. Accordingly, baseband processing block **210** may perform several operations such as analog-to-digital (A/D) conversion, pre-emphasis of channels **201**L and **201**R, generation of a mono-audio component (generated as the average of audio channels **201**L and **201**R), generation of a stereo-audio component (generated as the average difference of audio channels **201**L and **201**R), up-conversion of the stereo-audio component, inclusion of a pilot signal (for example, at 19 kHz), inclusion of data according to protocols such as Radio Data System (RDS) protocol, DARC (DAta Radio Channel), etc.

The term 'composite message signal' is used to refer to a combination of a mono audio component, a stereo audio component, a pilot signal and data according to protocols such as Radio Data System (RDS) protocol, DARC (DAta Radio Channel), etc. The term 'composite audio signal' will be used to refer to a combination of a mono audio component and a stereo audio component, without pilot signal and RDS components. FIG. **2**B shows the double-sided spectrum of a composite message signal **250**. Frequency band **251** represents the mono audio component (which includes left and right channels of an audio signal). Frequency band **252** represents the frequency shifted stereo audio component, the frequency shifting being performed with the aid of a pilot signal **255**. Frequency band **253** represents RDS data.

The composite message signal generated by baseband processing block **210** is sent to FM modulation block **220** on path **212**. FM modulation block **220** frequency modulates a carrier signal (not shown, but assumed to be provided to block **220** or generated within block **220**) with signal **212**, and provides a FM signal to power amplifier **230** on path **223**. Power amplifier **230** sends a power amplified FM signal on path **126**. The power amplified signal on path **126** is transmitted by an antenna (antenna **106** of FIG. **1**).

As is well-known in the relevant arts, pre-emphasis (that is performed in baseband processing block **210**) amplifies higher frequencies in each of audio channels **201**L and **201**R to a greater extent than lower frequencies in audio channels **201**L and **201**R, and may be performed consistent with corresponding FM standards. However, depending on the received amplitude levels (of the various frequency components of) of audio channels **201**L and **201**R, pre-emphasis may cause over-modulation in the FM signal, which may cause the FM modulated signal to occupy a wider bandwidth than may be allowed by the relevant FM standard(s).

Hence, it may be necessary to control the amplitude levels of frequency components (especially the higher frequency components) of the pre-emphasized audio signal so that the FM signal is not over-modulated.

2. Controlling Over-Modulation

FIG. **3** is a flow diagram illustrating the manner in which over-modulation in FM transmitters is controlled, in an embodiment. The steps in the flow diagram are described in a specific sequence merely for illustration. Alternative embodiments using a different sequence of steps can also be implemented without departing from the scope and spirit of several aspects of embodiments of the present disclosure, as will be apparent to one skilled in the relevant arts by reading the disclosure provided herein.

In step **310** of the flow diagram, a sequence of digital values representing an audio signal is formed. The sequence may represent a composite message signal, a composite audio signal, a stereo audio signal or a mono audio signal.

In step **320**, the deviation of the maximum amplitude level of the audio signal from a reference amplitude level is measured. The reference amplitude level represents a maximum frequency deviation limit, and may be determined a priori based, for example, on the correspondence between the amplitude of an audio signal and the bandwidth occupied by the corresponding FM signal.

In step **330**, the audio signal is filtered based on the measured deviation such that the gain (or attenuation) factors applied to different frequency components in the audio signal are different, depending on the frequency values of the components. A second sequence of digital values is generated as the output of the filtering of step **330**.

In step **340**, the second sequence of digital values is frequency modulated to generate a FM signal, which may then be transmitted.

As a result of the operations described above, frequency-dependent gain (or attenuation) factors are applied on the frequency components of an audio signal so that the bandwidth occupied by the FM signal is within the limits specified by the relevant FM standards, and over-modulation is controlled. Further, the operations described above are performed in the digital domain, and are performed using only digital blocks/components in baseband processing block **210** (shown in FIG. **2**). The operations of various blocks in a baseband processing block in controlling over-modulation are described next with respect to FIG. **4**.

3. Baseband Processing Block

FIG. **4** is a block diagram illustrating the details of a baseband processing block in an embodiment. Only the blocks as relevant to control of over-modulation are shown in FIG. **4**, and baseband processing block **400** of FIG. **4** may contain

more processing blocks as well. Baseband processing block **400** may be implemented in place of baseband processing block **210** of FIG. **2**, and is shown containing pre-emphasis filters **410**L and **410**R, variable de-emphasis filters **420**L and **420**R, summing blocks **430**, **440** and **460**, frequency shifters **450** and **465**, envelope detector **470**, log-level comparator **480** and de-emphasis control and hysteresis block **490**. The operations of the blocks of FIG. **4** may be performed in corresponding sampling intervals (or time index) of a clock. Baseband processing block **400** may be implemented using software, hardware, or a combination of hardware and software. In general, when throughput performance is of primary consideration, the implementation may be done more in hardware (e.g., in the form of an application specific integrated circuit). When cost is of primary consideration, the implementation may be performed more in software (e.g., using a processor executing instructions provided in software/firmware).

Pre-emphasis filters **410**L and **410**R perform pre-emphasis operations on left and right audio channels **210**L and **210**R respectively. As is well known in the relevant arts, pre-emphasis refers to an operation in which higher frequency components of an input audio signal are amplified (emphasized) more than lower frequency components of the audio signal (prior to frequency modulation of a carrier by the audio signal), and is generally performed to enable better signal-to-noise ratio (SNR) of the FM signal at a receiving end, i.e., at a FM receiver. Pre-emphasis operations in filters **410**L and **410**R may be performed consistent with requirements of relevant FM standards. In one embodiment, pre-emphasis filters **410**L and **410**R are implemented as Finite Impulse Response (FIR) filters. The (pre-emphasized) outputs of filters **410**L and **410**R are provided on paths **412**L and **412**R to variable de-emphasis filters **420**L and **420**R respectively.

Variable de-emphasis filter **420**L receives pre-emphasized signal **412**L, and filters signal **412**L based on input parameters β and δ, received respectively on paths **425** and **429**. Similarly, variable de-emphasis filter **420**R receives pre-emphasized signal **412**R, and filters signal **412**R based on input parameters β and δ. Parameter δ (which may be viewed as a control parameter) determines the attenuation (with respect to frequency) provided by each of filters **420**L and **420**R, and is generated/determined in a manner described below. Parameter β represents a time-constant parameter that modifies the attenuation provided by filters **420**L and **420**R for a given value of parameter δ.

Thus, filters **420**L and **420**R operate to perform frequency-dependent gain (or attenuation) control to alter the amplitude levels of frequency components of the respective pre-emphasized signals **412**L and **412**R. In general, the filters operate to attenuate higher frequency components to a greater extent than lower frequency components, thereby ensuring that over-modulation does not occur in FM transmitter.

In an embodiment, each variable de-emphasis filter (**420**L and **420**R) is implemented as a digital Infinite Impulse Response (IIR) filter whose impulse response is provided in equation 1 below:

$$H(z)=((1-\beta\delta)+\beta\delta z^{-1})((1-\delta)/(1-\delta z^{-1})) \quad \text{Equation 1}$$

wherein, H (z) represents the transfer function (or impulse response) of each of the filters.

Respective outputs **423**L and **423**R of variable de-emphasis filters **420**L and **420**R are provided to summing block **430**, which determines the average of signals **423**L and **423**R. The output of summing block **430**, referred to as mono-audio signal, is sent to summing block **460** on path **436** (mono). The band of frequencies occupied by signal **436** corresponds to frequency band **251** shown in FIG. **2**B.

Outputs **423**L and **423**R are also provided to summing block **440**, which determines the average difference between signals **423**L and **423**R. Stereo-audio signal **445** (Stereo) output by summing block **440** is frequency-shifted in frequency shifter **450** to a higher frequency band to spectrally separate signal **436** (mono) and **445** (stereo). An oscillator output (**455**) is used to mix the stereo-audio signal **445** (Stereo) to perform the frequency shift in a known way. The frequency of oscillator output **455** may be set as defined by the FM standards, and equals 38 KHz in an embodiment. The output of frequency shifter **450** is provided on path **456** to summing block **460**. Signal **456** represents the frequency-shifted stereo signal formed from left and right channels **201**L and **201**R. The band of frequencies occupied by signal **456** corresponds to frequency band **252** shown in FIG. **2**B.

Summing block **460** adds mono-audio signal **436** (mono) and frequency shifted stereo-audio signal on path **456** to form signal **461** containing both mono and stereo-audio signals. Frequency shifter **465** allows scaling of the amplitude of signal **461**, thereby allowing control of the extent of frequency deviation contribution of signal **461**. Frequency shifter **465** receives a scaling factor (audio frequency deviation factor) on path **462** to provide the scaling, and generates composite audio signal **467**.

Composite audio signal **467** may correspond to signal **212** of FIG. **2**. In other embodiments, RDS/DARC data may be added to signal **467** prior to further forwarding and processing (which may include the operations performed by blocks **220** and **230** of FIG. **2**), as well as those in blocks described below. Although in the description below blocks **470**, **480** and **490** are noted as operating based on composite audio signal **467** as input, in other embodiments signals **436** (mono) and/or signal **445** (stereo) may instead be provided as the input, with corresponding modifications in the operation of blocks **470**, **480** and **490**. In other embodiments, a composite message signal consisting of composite audio signal **467** and pilot **256** (FIG. **2**B) and/or RDS/DARC data can also be used as input to block **470**. In such alternative embodiments, the value of reference level **485** (described below) would be changed accordingly.

Envelope detector **470** may be implemented as a low pass filter and generates the envelope of signal **467**. Thus, envelope detector **470** computes the absolute value of the amplitude of the envelope in corresponding time (operation) intervals. The computed absolute value of the amplitude of the envelope is a measure of the frequency deviation that would be caused in a carrier signal that is modulated by signal **467** (for example in block **220** of FIG. **2**). In an embodiment, envelope detector **470** is implemented using a dual-mode single-pole low-pass IIR filter, and receives programmable time constants α_C (charge coefficient) and α_D (discharge coefficient) on path **475**. The absolute value of the amplitude of the envelope is determined according to the equations provided below:

$$l(n)=|x(n)|+\alpha_C(l(n-1)-|x(n)|),|x(n)|>l(n-1) \quad \text{Equation 2A}$$

$$l(n)=|x(n)|+\alpha_D(l(n-1)-|x(n)|),|x(n)|\leq l(n-1) \quad \text{Equation 2B}$$

wherein,

'n' represents a current sampling interval/time index,

l(n) represents the current sample at the envelope detector output,

x(n) represents the current sample at the envelope detector input,

l(n−1) represents the previous sample at the envelope detector output, and.

'||' represents an absolute-value operation.

Time constants α_C and α_D are used to ensure fast charging and slow-discharging behavior in envelope detector **470**. The output of envelope detector **470** is provided to log level comparator **480** on path **478**.

Log level comparator **480** converts the absolute values received on path **478** to a logarithmic scale, and compares the absolute values (in each of corresponding operation intervals of the circuit of FIG. **4**) with a reference amplitude value (which corresponds to a desired nominal frequency deviation), provided on path **485**. In an embodiment, the following equation represents the comparison performed by log level comparator **480**:

$$L(n)=\log_2(l(n)/I_{Nom})=\log_2(l(n))-\log_2(I_{NOM}) \quad \text{Equation (3)}$$

wherein,

L(n) represents the result of comparison noted above, and

I_{Nom} represents a reference amplitude level.

The sign of L(n) indicates the direction in which the de-emphasis response (lower or greater attenuation) of variable de-emphasis filters **420**L and **420**R is to be moved, i.e., whether the value of parameter δ has to be increased or decreased. The magnitude of L(n) indicates the amount by which the de-emphasis has to be changed, i.e., the magnitude change required in parameter δ. Log level comparator **480** provides the result L(n) to de-emphasis control and hysteresis block **490** on path **489**.

De-emphasis control and hysteresis block **490** varies the value of parameter δ depending on the output of log level comparator **480** received on path **489**, programmable hysteresis constants (DELTA_HYST_UP and DELTA_HYST_DOWN) received on path **495** and programmable time constants (μ_A and μ_R) received on path **496**. Parameter δ is varied according to the equations provided below:

$$\delta(n)=\mu_A L(n)+\delta(n-1), L(n)>0 \quad \text{Equation 4A}$$

$$\delta(n)=\mu_R L(n)+\delta(n-1), L(n)\leq 0 \quad \text{Equation 4B}$$

wherein,

μ_A and μ_R are coefficients, and

δ(n−1) is value of parameter δ at the previous sampling interval/time index.

Equations 4A and 4B, are referred to "update equations" for convenience, and indicate the manner in which the value of δ is "updated" (in a corresponding operation interval) depending on the value of L(n). Programmable time constants μ_A and μ_R are referred to as attack and release coefficients respectively. If the value of L(n) is positive and non-zero, then the coefficient μ_A is used in the update equation (4A), whereas μ_R is used otherwise (equation 4B). δ(n) is provided as a control input to filters **420**L and **420**R, and thus controls the attenuation provided by the filters. Coefficients μ_A and μ_R are selected to ensure fast attack and slow release, similar to the charge/discharge coefficients in the envelope detector.

It may be appreciated from the description above that parameter δ is dynamically varied using de-emphasis control and hysteresis block **490** to control the level of de-emphasis (that is, level of attenuation of higher frequency components of the audio signal) provided by the variable de-emphasis filters **420**L and **420**R. In an embodiment, control parameter δ is designed to have values in a range 0 to δ_{max}, with the value of δ_{max} corresponding to the frequency response of complete de-emphasis (i.e., the effect of pre-emphasis filters **410**L and **410**R on signals **201**L and **201**R respectively is completely negated). The value of δ_{max} is 1 in an embodiment. In general, the value of δ_{max} depends on the type of pre-emphasis specified in the relevant FM standard (i.e., 50 μS or 75 μS). Parameter β is used to vary the gain factors for different frequency components for a specific value of δ, and may potentially be set to different values for 50 μS and 75 μS de-emphasis, to tweak the response of filters **420**L and **420**R at δ_{max}, to better match the desired response of complete de-emphasis.

When δ equals zero, variable de-emphasis filters **420**L and **420**R act as all-pass filters, providing uniform gain (or no gain at all) to all frequency components of signals **412**L and **412**R. When parameter δ has a non-zero value, higher frequency components of the audio signal are attenuated to a greater extent than lower frequency components. The frequency response (variation of gain factors with respect to frequency components of audio signal) of variable de-emphasis filters **420**L and **420**R in an embodiment is depicted in FIG. **5**.

FIG. **5** depicts a set of curves, where each curve depicts the frequency response of variable de-emphasis filters **420**L and **420**R for a corresponding value of δ. The variation in frequency response corresponding to variation in the value of δ from 0 to δ_{max} is shown in FIG. **5**. Frequency (in KiloHertz) is represented on the X axis, and the magnitude (in decibels) of the corresponding frequency component is represented on the Y axis.

The manner in which control parameter δ is generated (implementation of update equations 4A and 4B) in de-emphasis control and hysteresis block **490** is briefly described next with reference FIG. **6**.

4. De-emphasis Control and Hysteresis

FIG. **6** is a block diagram illustrating the implementation blocks of a de-emphasis control and hysteresis block in an embodiment, and is shown containing hysteresis mechanism block **640**, multiplexers **620** and **621**, comparators **610** and **680**, minimum limiter **660**, maximum limiter **670**, multiplier **630**, summing block **631**, and unit time delay blocks **650** and **651**.

The value of L(n) received on path **489** from log-level comparator **480** (as described in FIG. **4**) is provided to comparator **610**. Comparator **610** checks if L(n) is greater than 0, and accordingly sends a signal on path **612**. Path **612** acts as a select line for multiplexer **620**. The two inputs to multiplexer **620** are paths **625** and **626**, carrying the values of attack and release coefficients μ_A and μ_R respectively. One of the two inputs (**625** and **626**) is selected depending on the value provided by comparator **610** on path **612**, and the selected coefficient (μ_A or μ_R) is provided to multiplier **630** on path **623**. Multiplier **630** multiples the values of L(n) and the selected coefficient (μ_A or μ_R) to generate a 'proposed update' σ(n), which is provided on path **634** to hysteresis mechanism block **640** and to multiplexer **621**.

Hysteresis mechanism block **640** uses proposed update σ(n) received on path **634**, and programmable hysteresis constants DELTA_HYST_UP and DELTA_HYST_DOWN received on paths **645** and **646** respectively, to generate a signal c(n) (provided on path **642** to multiplexer **621** as a select line). Paths **645** and **646** are deemed to be contained in path **495** of FIG. **4**.

The generated signal c(n) takes values based on the equation (5) provided below:

$$c(n)=1,$$

$$\text{if } \sigma(n)h(n)\geq 0 \text{ OR}$$

$$(h(n)=-1) \text{ AND } (\sigma(n)>\text{DELTA_HYST_UP}) \text{ OR}$$

$$(h(n)=+1) \text{ AND } (\sigma(n)<-\text{DELTA_HYST_DOWN})$$

$$c(n)=0, \text{ otherwise.} \quad \text{Equation 5}$$

wherein,

OR represents a logical OR operation,

AND represents a logical AND operation,

h(n) is a hysteresis state variable which represents the direction of previous changes in parameter δ.

The values of DELTA_HYST_UP and DELTA_HYST_DOWN may be chosen to mask small fluctuations in signal **489**, caused due to fluctuations in signal envelope **478**, which may occur in practical implementations of envelope detector **470**. In an embodiment, the values of DELTA_HYST_UP and DELTA_HYST_DOWN are 0.01 and 0.005 respectively.

h(n) is defined by equation (6) below.

$$h(n)=d(n-1) \text{ if } d(n-1)\neq 0$$

$$h(n)=h(n-1) \text{ if } d(n-1)=0 \qquad \text{Equation 6}$$

wherein,

d(n−1)=sign(δ(n−1)−δ(n−2))

h(0) is initialized to value 0.

Determination of d(n−1) used in equation 6 is implemented using unit time delay block **651** (from which the previous value of δ is obtained), comparator **680** (which compares the 'present' value of δ and the 'previous' value of δ to provide d(n) on path **685**), and unit time delay block **650** (which provides the value of d(n−1)). The value of d(n−1) generated by unit time delay block **650** is provided on path **654** to hysteresis mechanism block **640**, which uses d(n−1) to generate h(n), and further to generate c(n), which is provided to multiplexer **621** on path **642**.

Multiplexer **621** is used to prevent rapid fluctuations in the value of parameter δ (i.e., to provide hysteresis), by allowing the proposed update σ(n) to be used in the update equation only when c(n) received on select line **642** is 1. If c(n) has the value 1, multiplexer **621** sends the update term σ(n) to summing block **631** on path **623**, and zero otherwise. Summing block **631** adds update term σ(n) to the previous value of parameter δ to generate a new value for parameter δ. The previous value of δ (that is, δ (n−1)) is obtained from unit time delay block **651** on path **653**. The generated δ value is provided to minimum limiter **660**.

Minimum limiter **660** checks the generated value of δ against a minimum value δ_{min} (which may equal 0, although the specific value is programmable) received on path **665**, and chooses the maximum of the two values to be the new δ value (that is, limiting the lowest value of δ to δ_{min}). The output of minimum limiter **660** is provided to maximum limiter **670**.

Maximum limiter **670** checks the value of δ received on path **667**, against a maximum value δ_{max} received on path **675**, and chooses the minimum of the two values to be the new δ value (limiting the highest value of δ to δ_{max}). Minimum limiter **660** and maximum limiter **670** in conjunction act as a range control mechanism, limiting the generated value of δ between δ_{min} and δ_{max}.

The range-limited δ value is provided on path **492** (shown in FIG. **4**) to variable de-emphasis filters **420**L and **420**R, and is used by filters **420**L and **420**R to set their impulse response. Future outputs of filters **420**L and **420**R (i.e., outputs after application of the δ value on path **492**) result in generation of a new sequence of digital values representing composite audio signal **467** (FIG. **4**), with the mono and stereo-audio signals having their audio frequency components de-emphasized to ensure that over-modulation is controlled. The 'new' sequence of digital values is provided on path **212** (corresponding to output of baseband processing block **200** in FIG. **2**) to FM modulation block **220**, in which a signal represented by the 'new' sequence of digital values is frequency modulated.

The frequency dependent de-emphasis (gain control) described above limits the bandwidth occupied by a FM signal ensuring that the spectral-mask requirements imposed by relevant FM standards are met. Since the technique described above is implemented completely in the digital domain, it may be easily integrated in environments which are primarily digital, such as mobile phone **100** of FIG. **1** (in general, in low-power, small-area FM transmitters). In addition, the provision of the same value of control parameter δ to both of filters **420**L and **420**R ensures matching in the filtering applied on audio channels **201**L and **201**R, thereby ensuring no impact on stereo separation.

Although in the foregoing description, the generation of parameter δ is described as being performed based on measurement of amplitude of composite audio signal **467**, in other embodiments δ may be generated based on independent measurement of audio channels **201**L and **201**R. Correspondingly, a pair of δ values may then be generated and applied independently as control parameters for setting the impulse response of filters **420**L and **420**R.

References throughout this specification to "one embodiment", "an embodiment", or similar language mean that a particular feature, structure, or characteristic described in connection with the embodiment is included in at least one embodiment of the present disclosure. Thus, appearances of the phrases "in one embodiment", "in an embodiment" and similar language throughout this specification may, but do not necessarily, all refer to the same embodiment.

While various embodiments of the present disclosure have been described above, it should be understood that they have been presented by way of example only, and not limitation. Thus, the breadth and scope of the present disclosure should not be limited by any of the above-described embodiments, but should be defined only in accordance with the following claims and their equivalents.

What is claimed is:

1. Frequency modulated baseband circuitry comprising:

(A) left audio channel pre-emphasis filter circuitry having an output, and right audio channel pre-emphasis filter circuitry having an output;

(B) first summing circuitry having a first input coupled to the output of the left audio channel pre-emphasis filter circuitry, having a second input coupled to the output of the right audio channel pre-emphasis filter circuitry, and having an output;

(C) first difference circuitry having a first input coupled to the output of the left audio channel pre-emphasis filter circuitry, having a second input coupled to the output of the right audio channel pre-emphasis filter circuitry, and having an output;

(D) second summing circuitry having a first input coupled to the output of the first summing circuitry, a second input coupled to the output of the first difference circuitry, and an output;

(E) frequency shifter circuitry having an input coupled to the output of the second summing circuitry, a second input receiving an audio frequency deviation factor, and a composite audio signal output;

(F) left de-emphasis filter circuitry having an input coupled to the output of the left audio channel pre-emphasis filter circuitry, having a control parameter input coupled to the composite audio signal output, and an output coupled to the first inputs of the first summing circuitry and the first difference circuitry; and

(G) right de-emphasis filter circuitry having an input coupled to the output of the right audio channel pre-emphasis filter circuitry, having a control parameter input coupled to the parameter input of the left variable de-emphasis filter circuitry, and an output coupled to the second inputs of the first summing circuitry and the first difference circuitry.

2. The circuitry of claim 1 including:

(i) envelope detector circuitry having an input coupled to the composite audio signal output and an output;

(ii) log level comparator circuitry having an input coupled to the output of the envelope detector circuitry and having an output; and

(iii) control circuitry having an input coupled to the output of the log level comparator circuitry and having a control parameter output coupled to the control parameter input of the left de-emphasis filter circuitry.

3. The circuitry of claim 1 including envelope detector circuitry having an input coupled to the composite audio signal output, a time constant input, and an output coupled to the control parameter input of the left de-emphasis filter circuitry.

4. The circuitry of claim 1 including log level comparator circuitry having an input coupled to the composite audio signal output, a reference level input, and an output coupled to the control parameter input of the left de-emphasis filter circuitry.

5. The circuitry of claim 1 including control circuitry having an input coupled to the composite audio signal output, a hysteresis input, a time constant input, and an output coupled to the control parameter input of the left de-emphasis filter circuitry.

6. The circuitry of claim 1 in which the left and right de-emphasis filter circuitry each have a time constant parameter input connected together in addition to the control parameter input.

7. A process of producing a composite audio signal to a frequency modulation circuit comprising:

(A) pre-emphasis filtering a left channel audio signal separately from pre-emphasis filtering a right channel audio signal;

(B) de-emphasis filtering the pre-emphasis filtered left channel audio signal in response to a control parameter separately from de-emphasis filtering the pre-emphasis filtered right channel audio signal in response to the control parameter;

(C) producing a composite audio signal to a frequency modulation circuit from the de-emphasis filtered left channel audio signal and the de-emphasis filtered right channel audio signal; and

(D) producing the control parameter from the composite audio signal.

8. The process of claim 7 in which producing the control parameter from the composite audio signal includes:

(i) producing an envelope detector output signal from the composite audio signal;

(ii) producing a log level comparison output signal from the envelope detector output signal;

(iii) producing the control parameter output from the log level comparator output signal.

9. The process of claim 7 in which producing the control parameter from the composite audio signal includes producing an envelope detector output signal from the composite audio signal and a time constant input signal.

10. The process of claim 7 in which producing the control parameter from the composite audio signal includes producing a log level comparison output signal from the composite audio signal and a reference level input signal.

11. The process of claim 7 in which producing the control parameter from the composite audio signal includes producing an the control parameter output from the composite audio signal, a hysteresis input signal, and a time constant input signal.

12. The process of claim 7 in which de-emphasis filtering the pre-emphasis filtered left and right channel audio signals includes de-emphasis filtering in response to a time constant parameter signal.

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